12-27-05

AF
IFW

Express Mail Label No.: EV 532353935 US
Date of Deposit: December 23, 2005

Attorney Docket No. 27996-097

OIPE IAP64
DEC 2 3 2005
PATENT & TRADEMARK OFFICE

BEST AVAILABLE COPY

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Appl. No.	:	09/750,766	Confirmation No.	8660
Applicant(s)	:	Brian B. Eagan		
Filed	:	December 28, 2000		
TC/A.U.	:	2665		
Examiner	:	Clemence S. Han		
Docket No.	:	27996-097		
Customer No.	:	35437		
Title	:	VOICE OPTIMIZATION IN A NETWORK HAVING VOICE OVER INTERNET PROTOCOL COMMUNICATION DEVICES		

Mail Stop Appeal Brief-Patents-Fee
Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

TRANSMITTAL LETTER

Transmitted herewith for filing in the above-identified patent application are the following documents:

1. Appellants' Brief Pursuant to 37 C.F.R. § 1.192; (13 pgs);
2. Check No. 2983 in the amount of $500.00; and
3. Return postcard.

The Commissioner is hereby authorized to charge any fee that may be due, or to credit any overpayment, to Deposit Account No. **50-0311,** Ref. No.: **27996-097**. A duplicate copy of this transmittal letter is enclosed.

Respectfully submitted,

Dated: December 23, 2005

Boris A. Matvenko, Reg. No. 48,165
Attorney for Applicants
MINTZ, LEVIN, COHN, FERRIS
GLOVSKY AND POPEO, P.C.
Chrysler Center, 666 Third Ave., 24th Fl.
New York, NY 10017
Tel: (212) 935-3000
Fax: (212) 983-3115

Express Mail Label No.: EV 532353935 US
Date of Deposit: December 23, 2005

Attorney Docket No.: 27996-097

OIPE IAP64 DEC 2 3 2005 PATENT & TRADEMARK OFFICE

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Applicant(s):	Brian B. Eagan	**Examiner:**	Han, C.S.
Application No.:	09/750,766	**Art Unit:**	2665
Filing Date:	December 28, 2000	**Confirmation No.:**	8660
		Atty Docket No.:	27996-097

Title: **VOICE OPTIMIZATION IN A NETWORK HAVING VOICE OVER INTERNET PROTOCOL COMMUNICAITON DEVICES**

Mail Stop Appeal Brief-Patents - FEE
Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

APPEALLANTS' BRIEF PURSUANT TO 37 C.F.R. § 1.192

In accordance with a Notice of Appeal, filed on November 10, 2005, Applicants submit this Appellants' brief.

1. **Fee**: Enclosed herewith is a check for the fee of $500.00 for filing of a brief in support of an appeal.

2. **Real Party-in-Interest**: All rights to the above referenced patent application have been assigned to:

Nortel Networks Limited
2351 Boulevard Alfred-Nobel
St. Laurent, Quebec H4S 2A9, CANADA

3. **Related Appeals and Interferences**: There are no known other appeals or interferences that would directly or indirectly affect the Board's decision in the present appeal.

12/28/2005 CCHAU1 00000004 09750766
01 FC:1402 500.00 OP

4. **Status of the Claims of U.S. Patent Application Serial No. 09/750,766 ("'766 application")**:

Claims 1-24 are pending.

Claims 1-6, 10-18, and 20-23 stand rejected under 35 U.S.C. 102(e) as being anticipated by U.S. Patent No. 6,700,895 to Kroll ("Kroll").

Claims 7-9, 19, and 24 are objected as being dependent upon a rejected base claims, but would be allowable if rewritten in independent form including all of the limitation of the base claim and any intervening claims.

5. **Status of Amendments**:

(a) A First Office Action was mailed June 7, 2004.

(b) A Response to the First Office Action was filed October 7, 2004, traversing the Examiner's rejections and objections.

(c) A Second Office Action was mailed February 25, 2005.

(d) A Response to the Second Office Action was filed May 23, 2005, traversing the Examiner's rejections.

(e) A Final Office Action was mailed August 10, 2005.

(f) An Interview with the Examiner was conducted on September 20, 2005, again traversing the Examiner's rejections.

(g) Notice of Appeal was filed November 10, 2005.

(h) No amendment was filed after the Final Office Action.

6. **Summary of the Claimed Subject Matter**:

The present invention is directed to an apparatus and a method for optimizing voice quality on networks that employ voice over Internet Protocol ("VoIP") communication devices.

The optimization is based on measuring network performance pertaining to a quality of the network connection and evaluating the same. (Page 1, lines 28-32; Page 3, lines 21-23).

To optimize voice quality on the network containing VoIP end-point devices, the default parameters of at least some of these devices are initialized and later adjusted, as necessary. (FIGS. 1 and 3; Page 6, lines 13-15). The parameters include CODEC selection, packet size (a number of frames per packet), desired latency, packet loss, available bandwidth, number of router hops, and jitter buffer size. (Page 6, lines 24-28).

At the time of initialization, the VoIP end-point devices register with private branch exchange ("PBX"). In response to registration, a terminal proxy server uses a protocol to instruct the end-point devices to use a CODEC of a particular type, a jitter buffer of a particular size, a frame size of so many voice samples, etc. (FIG. 1; Page 7, lines 6-11).

When a VoIP end-point device registers with an internet protocol PBX ("IP PBX"), the IP PBX may perform a number of tests to determine the optimum configuration for that end-point. A number of software tools can be used to measure network performance (e.g., jitter, available bandwidth, delay, packet loss, latency, etc.), which can be reevaluated after initialization. (Page 6, line 30 to Page 7, line 5; Page 7, lines 22-24). The software tools include a ping tool, a network trace tool and a packet loss measurement tool. (Page 6, line 34 to Page 7, line 1).

The present invention applies a three-phase approach to optimize quality of a VoIP connection: initializing, network performance monitoring/measurement and dynamically intervening or correcting. (FIG. 2; Page 8, line 33 to Page 9, line 1). The initialization and the network performance monitoring/measurement phases are described above. As a result of the network performance monitoring/measurement, the default parameters for the end-point devices

are changed when the measured performance parameters signify that a connection to the network is below a desired level of operation. (Page 9, lines 16-18). The terminal proxy server and the end-point devices make the necessary measurements using the above software tools and report findings to the terminal proxy server, which uses conventional protocol to instruct the end-point devices to make appropriate changes in their default parameter settings. (Page 9, lines 27-31).

The three-phase approach gives the best audio quality with the lowest amount of latency and uses the minimum amount of digital signal processor resources. (Page 10, lines 29-31). The approach optimizes voice quality for the IP networks that have and have not been provisioned for quality of service (QoS). (Page 12, lines 4-9).

The Applicants note that claims 1-24 stand and fall together. However, as required by MPEP 1205, 37 C.F.R. 41.37(c)(1)(v), the Applicants provide herewith specification reference points for each element in independent claim 20. The Applicants note that these reference points are for exemplary purposes only and are not intended to limit the scope of claim 20.

At least Page 1, lines 28-32 and Page 3, lines 21-23 of the present application's specification describe the preamble of claim 20 that states: "An apparatus to effect voice optimization in a packet switched network." At least FIGS. 1 and 3; Page 6, lines 13-15 and lines 24-28 describe "means for initializing default parameters for end-point devices on a network with respect to choice of preferred CODEC, number of voice samples per packet, and jitter buffer size." At least FIG 1; Page 6, line 30 to Page 7, line 5; and Page 7, lines 6-11 and 22-24 describe "means for measuring performance parameters of a network." At least FIGS. 1 and 2; Page 6, line 30 to Page 7, line 5; Page 8, line 33 to Page 9, line 1; and Page 9, line 16-18 and 27-31 of the specification describe "means for making a determination as to whether the measured performance parameters signify that a connection to the network is below a desired

level of operation." At least FIGS. 1-3; Page 6, lines 13-15 and lines 24-28; Page 8, line 33 to Page 9, line 1; Page 9, line 16-18 and 27-31 describe "means for adjusting the default parameters based upon the determination being that the measured performance parameters signify that the connection to the network is below the desired level of operation."

7. **Grounds of Rejection to be Reviewed on Appeal**:

Applicants contend that claims 1-24 are novel and not anticipated under 35 U.S.C. 102(e) by Kroll and claims 7-9, 19 and 24 do not depend from invalid base claims.

8. **Argument**:

Independent claims 1, 14 and 20 are not anticipated under 35 U.S.C. 102(e) by Kroll.

In the Final Office Action, the Examiner cited Kroll as an anticipating reference with respect to claims 1-6, 10-18 and 20-23.

Claim 1 of the present application recites, *inter alia*, a method of voice optimization in a packet switched network, comprising: initializing default parameters for end-point devices on a network with respect to choice of preferred CODEC, number of voice samples per packet, and jitter buffer size.

Kroll discloses a method and a system for computationally efficient calculation of frame loss rates over an array of virtual buffers. (See, Abstract). Kroll predicts the frame loss rates of arbitrary buffer sizes based upon the arrival statistics for a sample of the packet stream through the network channel. (Col. 3, lines 63-66). Kroll's jitter buffer size can be selected from multiple sizes. (Col. 7, lines 10-18). Kroll calculates a desired frame loss rate based on the selected buffer size. Once, the frame loss is computed for each virtual buffer, an optimal buffer size is then chosen that provides a desirable amount of jitter compensation. (Col. 7, lines 27-32).

As admitted by the Examiner (See, August 10, 2005 Office Action, Page 4, Para. 4) and as pointed out by the Applicant, Kroll does not disclose initializing a preferred CODEC or a number of voice samples per packet. In the August 10, 2005 Office Action, the Examiner implies that claim 1 is a Markush claim and thus it is possible to invalidate the claim by finding only one of the listed parameters for initialization. Accordingly, the Examiner concluded that because Kroll discloses jitter buffer size, it anticipates claim 1.

Clearly, claim 1 is not a Markush claim. According to MPEP,"...[a] Markush-type claim recites alternatives in a format such as 'selected from the group consisting of A, B and C.' See *Ex parte Markush*, 1925 C.D. 126 (Comm'r Pat. 1925)." (See, MPEP 803.02). Claim 1 does not recite initialization of one parameter from a group of parameters; instead, it requires initialization of all three parameters, i.e., preferred CODEC, number of voice samples per packet, and jitter buffer size.

By way of an example, claim 2 is a Markush claim. Claim 2 contains a phrase "from a group consisting of," which indicates that a selection of one of the elements in the group is desired. This is different from the language used in claim 1, where all three elements are required for initialization. Kroll does not disclose initialization of all three parameters. Hence, it does not disclose all elements of claim 1 and claim 1 should be allowed. Claims 2-13 depend from claim 1 and are not anticipated by Kroll for at least the reasons stated above with respect to claim 1.

Independent claims 14 and 20 include limitations that are the same as or similar to those of independent claim 1 and, thus, are not anticipated by Kroll for least the same reasons. Claims 15-19 and 21-24 depend from claims 14 and 20, respectively, and are allowable over Kroll for at least the reasons stated above with respect to claim 1. Hence, the objection of claims 7-9, 19 and 24 is now moot.

CONCLUSION

All pending claims of the application are valid over the cited references. Allowance of the application is respectfully requested.

Dated: December 23, 2005

Respectfully submitted,

Boris A. Matvenko, Reg. No. 48,165
MINTZ, LEVIN, COHN, FERRIS,
GLOVSKY & POPEO, P.C.
Chrysler Center, 666 Third Avenue
New York, NY 10017
Tel: (212) 935-3000
Fax: (212) 983-3115

APPENDIX A

Copy of Claims

1. (Original) A method of voice optimization in a packet switched network, comprising:

initializing default parameters for end-point devices on a network with respect to choice of preferred CODEC, number of voice samples per packet, and jitter buffer size;

measuring performance parameters of a network; and

evaluating whether the measured performance parameters signify that a connection to the network is below a desired level of operation and, if so, adjusting the default parameters for the end-point devices based on the evaluating.

2. (Original) A method as in claim 1, wherein the adjusting includes performing functions that are selected from a group consisting of re-negotiating a CODEC connection, re-setting of parameters for the packet size and re-setting the jitter buffer.

3. (Original) A method as in claim 2, wherein the performance parameters being measured are selected from a group consisting of throughput, latency, packet loss, bandwidth, number of network hops to the end-point devices, round trip delay and any combination thereof.

4. (Original) A method as in claim 3, wherein the measuring is performed with at least one tool selected from a group consisting of a ping tool, a network trace tool and a packet loss measurement tool.

5. (Original) A method as in claim 1, wherein the performance parameters being measured are selected from a group consisting of throughput, latency and packet loss, bandwidth, number of network hops to the end-point devices, round trip delay, and any combination thereof.

6. (Original) A method as in claim 5, wherein the measurements are obtained from measuring with at least one tool selected from a group consisting of a ping tool, a network trace tool and a packet loss measurement tool.

7. (Original) A method as in claim 1, wherein the adjusting is manually initiated by a user.

8. (Original) A method as in claim 2, wherein the adjusting is manually initiated by a user.

9. (Original) A method as in claim 1, further comprising registering the end-point devices with a private branch exchange (PBX) on the network, wherein said PBX measures performance parameters between the PBX and the end-point to determine the default parameters.

10. (Previously Presented) A method as in claim 1, further comprising:

measuring and evaluating existing performance parameters with respect to quality of connection, the initializing being based on the evaluating.

11. (Previously Presented) A method as in claim 10, wherein the existing performance parameters being measured are selected from a group consisting of throughput, latency, packet loss, bandwidth, number of network hops to the end-point devices, round trip delay and any combination thereof.

12. (Previously Presented) A method as in claim 1, further comprising evaluating the measured performance parameters with respect to quality of connection and performing the adjusting as a result of the evaluating.

13. (Previously Presented) A method as in claim 1, wherein the adjusting is carried out during transmission of media to the end-point devices.

14. (Previously Presented) An apparatus to effect voice optimization in a packet switched network, comprising:

an initializer configured and arranged to initialize default parameters for end-point devices on a network with respect to choice of preferred CODEC, number of voice samples per packet, and jitter buffer size;

a measurer configured and arranged to measure performance parameters of a network;

an evaluator configured and arranged to make a determination as to whether the measured performance parameters signify that a connection to the network is below a desired level of operation; and

an adjuster configured and arranged to adjust the default parameters based upon the determination being that the measured performance parameters signify that the connection to the network is below the desired level of operation.

15. (Previously Presented) An apparatus as in claim 14, wherein the measurer includes software tools configured to measure the performance parameters, the performance parameters being selected from a group consisting of throughput, latency, packet loss, bandwidth, number of network hops to the end-point devices on the network, round trip delay and any combination thereof.

16. (Previously Presented) An apparatus as in claim 15, wherein the measurer includes software tools configured to measure the performance parameters, the software tools including at least one tool selected from a group consisting of a ping tool, a network trace tool and a packet loss measurement tool.

17. (Previously Presented) An apparatus as in claim 16, wherein software tools include at least one tool selected from a group consisting of a ping tool, a network trace tool and a packet loss measurement tool.

18. (Previously Presented) An apparatus as in claim 14, wherein the adjuster is configured and arranged to perform functions which are selected from a group consisting of a re-negotiation of a CODEC connection and a re-set of the default parameters for the packet size and a re-set of the default parameters for jitter buffer size.

19. (Previously Presented) An apparatus as in claim 14, further comprising a private branch exchange on the network; a register configured to register the end-point devices with the private branch exchange (PBX) on the network; and a controller responsive to the register completing registration of the end-point devices with the PBX to direct the initializer to initialize the default parameters.

20. (Previously Presented) An apparatus to effect voice optimization in a packet switched network, comprising:

means for initializing default parameters for end-point devices on a network with respect to choice of preferred CODEC, number of voice samples per packet, and jitter buffer size;

means for measuring performance parameters of a network;

means for making a determination as to whether the measured performance parameters signify that a connection to the network is below a desired level of operation; and

means for adjusting the default parameters based upon the determination being that the measured performance parameters signify that the connection to the network is below the desired level of operation.

21. (Previously Presented) An apparatus as in claim 20, wherein the measuring means includes software tools configured to measure the performance parameters, the performance parameters being selected from a group consisting of throughput, latency, packet

loss, bandwidth, number of network hops to the end-point devices on the network, round trip delay and any combination thereof.

22. (Previously Presented) An apparatus as in claim 20, wherein the measuring means includes software tools configured to measure the performance parameters, the software tools including at least one tool selected from a group consisting of a ping tool, a network trace tool and a packet loss measurement tool.

23. (Previously Presented) An apparatus as in claim 20, wherein the adjusting means includes means for re-negotiating a CODEC connection, means for re-setting the default parameters for the packet size and means for re-setting the default parameters for the jitter buffer size.

24. (Previously Presented) An apparatus as in claim 20, further comprising a private branch exchange on the network; means for registering the end-point devices with the private branch exchange (PBX) on the network; and means responsive to the registering means completing registration of the end-point devices with the PBX for directing the initializing means to initialize the default parameters.